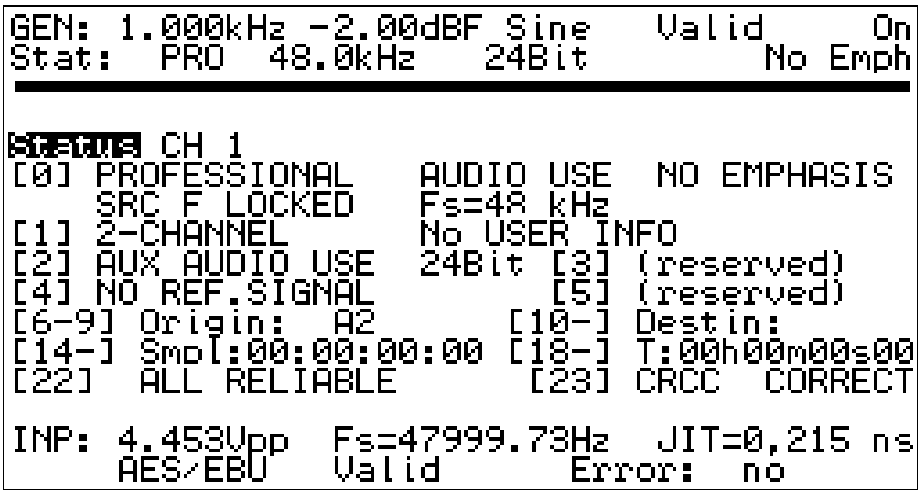
User Manual

AO10

Digital Extension for the

A2

Audio Test & Measurement System


GEN: 1.000kHz -2.00dBF Sine Valid On
Stat: PRO 48.0kHz 24Bit No Emph
Status CH 1
[0] PROFESSIONAL AUDIO USE NO EMPHASIS
SRC F LOCKED Fs=48 kHz
[1] 2-CHANNEL No USER INFO
[2] AUX AUDIO USE 24Bit [3] (reserved)
[4] NO REF.SIGNAL [5] (reserved)
[6-9] Origin: A2 [10-] Destin:
[14-] Smpl:00:00:00:00 [18-] T:00h00m00s00
[22] ALL RELIABLE [23] CRCC CORRECT
INP: 4.453Vpp Fs=47999.73Hz JIT=0,215 ns
AES/EBU Valid Error: no

Version 5.1
A2 Firmware V2.46 or higher
For all A2 with S/No. 400 up

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- pro audio with a smile

INTERNATIONAL WARRANTY

Limited Warranty

NEUTRIK guarantees the A2-D Audio Test & Service System and its components against defects in material or workmanship for a period of one year from the date of original purchase, and agrees to repair or to replace any defective unit at no cost for either parts or labour during this period.

Restrictions

This warranty does not cover damages that have resulted of accidents, misuse, lack of care, the attachment or installation of any components that were not provided with the product, loss of parts or connecting the instrument to any other power supply, input signal voltage or connector type than specified. In particular, no responsibility is granted for special, incidental, or consequential damages.

This warranty becomes void if servicing or repairs of the product are performed by any other party than an authorized service center.

No other warranty, written or oral, is authorized by NEUTRIK. Except as otherwise stated in this warranty, NEUTRIK makes no representation or warranty of any kind, expressed or implied in law or in fact, including, without limitation, merchasing or fitting for any particular purpose and assumes no liability, either in tort, strict liability, contract or warranty for products.

NOTE

In case of malfunction, take - or ship prepaid - your NEUTRIK A2-D Audio Test & Service System, packed in the original packing, to your nearest authorized service center. Be sure to include a copy of your sales invoice as proof of purchase date. Transit damages are not covered by this warranty.

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1 INTRODUCTION

The AO10 digital option is an extremely useful and versatile extension for the A2 Audio Measurement System. It offers all necessary functions to reliably test AES/EBU signals, connections or functions, analyzed through balanced, unbalanced or optical links: Nevertheless, the operation of all these measurements is as simple of the analog part.

In paractice, when working in a mixed signal environment, manifold problems may arise: non-matching sampling rates, format incompatibilities, impedance mismatches, jitter being introduced by poor I/O circuits or clock distribution problems etc. All of these problems may cause complex failures, ranging from occasionally audible effects to very subtle distortions or even failing systems.

Fortunately, the AO10 extension allows the A2-D Measurement System to operate in the analog, digital or even mixed mode. The option provides a sophisticated interface to synchronize and handle the AES/EBU and IC958 (SPDIF) formats, as well as a versatile digital generator, allowing to feed the DUT with all necessary test signal parameters.

Generator

The digital generator supports AES/EBU (AES-3) and IEC958 signals. The provided standard *Sampling Frequencies* are 32kHz, 44.1kHz and 48kHz. Additionally, the oscillator may also be triggered by the analyzer input or by an external clock.
The generator supports a **Sine** signal in the **PRO**fessional or **Consum**er format with a *Resolution* from 4bit to 24bits. In the **Pass** mode, the analyzer input signal is looped through the unit to the generator output, while the status or amplitude of the signal may be modified. The built-in **Jit**ter feature allows to introduce a defined jitter for acceptance testing.

Built-in sweep capabilities allow e.g. the quick & precise qualification of A/D and D/A converters, using the combination of analog generation and digital analysis and vice versa.

AES/EBU signals are brought to/from the unit through the XLR connectors at the front panel, where the system automatically detects the signal type (analog or digital) and enables the corresponding analysis mode. RCA and optical TOSLINK connectors are provided at the rear panel, together with the input clock and the trigger output BNC connectors.

Analyzer

The analyzer continuously measures the *Sampling Frequency* as well as the *Carrier* peak amplitude, thus indicating cable loss and impedance mismatching problems. The clock for the D/A or A/D conversion is normally derived from the input signal itself.
Since word clock jitter is responsible for poor audio data, the AO10 continuously measures this type of *Jitter* in two ranges - a wider range for the less accurately specified consumer products and a very precise range with a resolution of less than 1ns for professional equipment.

An on-line *Status Bit* analysis covers the contents of the status and the auxiliary bits, which are displayed in binary and also as decoded information. Transmission problems like parity errors, lacking confidence, validity and lock mode errors are recognized and displayed. The *Bit Statistics* easily detects hanging bits in any position of the word, including the audio data and AUX bits to quickly find failing bits.

A high performance D/A converter enables the user to monitor the coded audio signal to identify lines and channels. An internal link from the converted signal to the analyzer of the

A2-D allows to do all the conventional analog audio analysis with the digital signals, e.g. measurements of the *RMS Level* or the *THD+N* (distortion) of the signals. *Peak Levels* can be measured digitally and are expressed either in dBF (dB Full scale) or percentage (%). A dual-channel analog bargraph complements the result presentation in real-time.

The SYNC output provides selectable trigger signals of either the X-, Y- or Z- preamble.

The FFT option (AO12) further enhances the analyzing abilities of the system by introducing a new display mode for the analysis of analog input signals in the frequency domain. Actually, the FFT feature provides a real time spectral analysis with a high dynamic range from 0Hz-5kHz or 0Hz-20kHz, thus adding a real time spectrum analyzer (RTA) to the system.

AO10 Block Diagram

The AO10 digital option can be regarded as an independent analyzer / generator board with switchable in- & outputs. An internal link to the analog analyzer of the A2 through the D/A converter allows to make the digital input signal audible and to analyze it with the standard analog measurement functions.

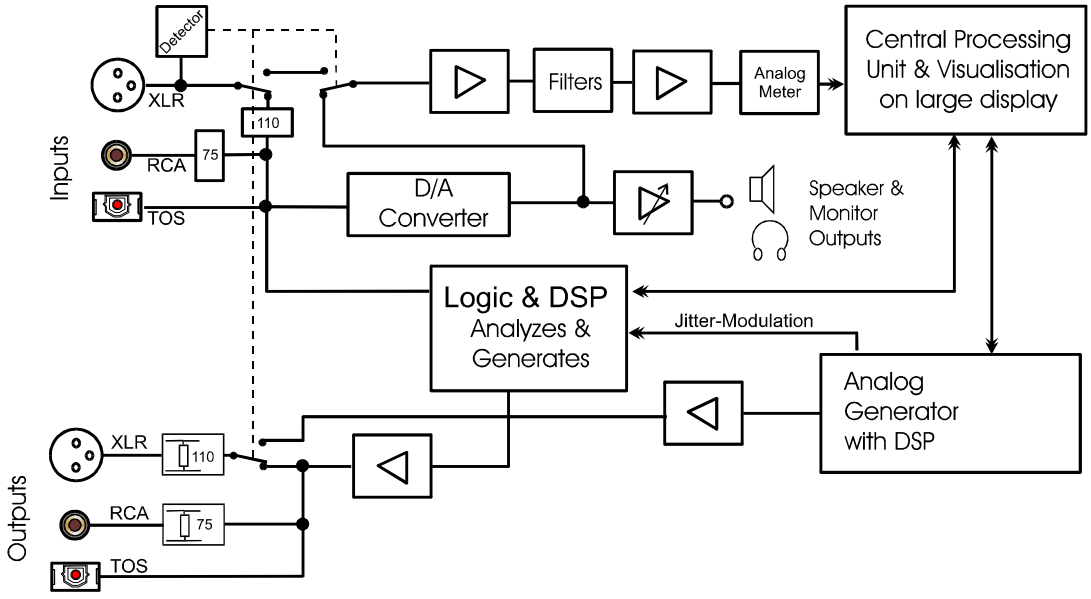


Fig. 1 Block Diagram

The detector behind the XLR input connector automatically detects the signal type and activates the corresponding mode. Digital analysis is performed in the DSP itself. The signal is parallel applied to the D/A converter for acoustical monitoring. Generation of AES/EBU signals is also performed by the DSP. For jitter generation the analog generator is used as the jitter source providing different waveforms, frequency and amplitude.

Connectors Overview

AES/EBU signals are normally connected through XLR connectors that are terminated with 110Ω, while the IEC958 signals (SP-DIF) are connected through RCA or optical TOSLINK connectors with 75Ω termination.

AES/EBU in- & outputs are available at the front panel of the A2-D. The XLR connectors of CH A (and only CH A) are prepared to handle both digital and analog signals. Due to the fact, that a 2-channel digital audio signal is encoded in a serial bit stream, it is transmitted through one cable, thus requiring only one I/O connector each. Please refer also to chapter ***7 AES/EBU Definitions***.
The RCA and optical TOS link connections are provided at the rear panel of the A2-D. They are marked with INPUT and OUTPUT.
The digital output signal is always available simultaneously at all three output connectors.

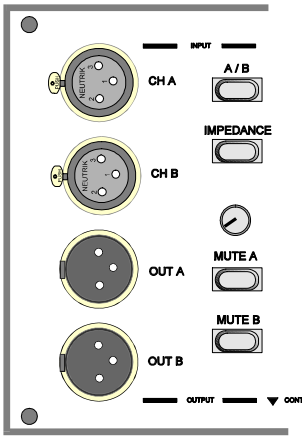


Fig. 2 AES 3 I/O

On the other hand, the three input connectors (XLR, RCA and SP-DIF) are not connected in parallel in order to avoid unwanted interferences between the possibly different input clock signals. Therefore, it is necessary to select the active input connector (please refer to chapter ***Input Connector*** (p. ***33***). Furthermore, it is strongly recommended never to connect a digital signal line to more than one input connector of the A2-D.

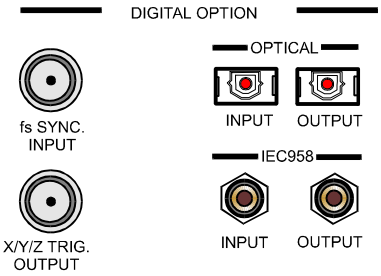


Fig. 3 SP-DIF I/O

fs Sync Input

Basically, there are three different clock rates defined for digital audio signals: 32kHz, 44.1kHz and 48kHz.
Additionally, the A2-D generator may be synchronized to an external clock with up to 52kHz. To do this, connect the corresponding clock signal to the *fs SYNC INPUT* on the rear panel of the unit. Please notice, that this mode does not allow to de-tune the output clock frequency or to create any jitter as described in chapters ***Fs*** (p. ***19***) and ***Jitter Amplitude***.

X/Y/Z Trigger Output

Alternatively, the A2-D is also able to act as a master, by providing different types of output clocks. The available clock signals are: X-, Y-, and Z-preamble clock as well as the bit clock for the eye pattern test.

Monitor / Headphone Output

Like in the analog mode, the headphone output at the rear panel provides the (D/A converted) audio signal of the selected digital input channel. In case of the THD+N measurement function, the reading signal (i.e. the harmonics + noise) are provided at the headphone jack.
The headphone output signal is also routed to the internal loudspeaker of the A2-D, so that the signal may be monitored without a headphone set as well.

How to Use this Manual

This User Manual has been designed with you, the operator in mind, to serve as a complete reference document for using the digital part of the A2-D as a solution to your measurement tasks.

Familiarize yourself with the A2-D by looking through this manual. The best way to feel at ease with the instrument is to sit down with the A2-D in front of you, and to start with the next chapter. By this way, it won't take long to learn to know the instrument and its many features.

Conventions

Following conventions were made in this manual in order to simplify the correlation to the user interface of the A2-D.

- <KEYS> and <SOFTWHEELS> are indicated by capital letters between pointed brackets.
- The *Names* of settable parameters as well as *Menus* / *Menu-Entries* are in italic.
- **Entry-** and **Result-Fields** in the digital LC display mode and are written in bold letters.

2 OPERATION

Activation of the Digital Analyzing Mode

Manual Activation

If the A2-D operates in the anlog mode, it can be switched over to the digital analyzing mode by pressing the <*ADDITIONAL*> key, which is the second function of the <PHASE> button. Every keypress to this button toggles the analyzer between the analog Phase measurement function and the digital analysis mode.
To return to the analog mode, make sure that no digital signal is present at the XLR input connector, and enter any of the analog measurement functions by pressing the corresponding key.
The mixed signal analysis is explained in the chapter ***Mixed Signal Analysis***.

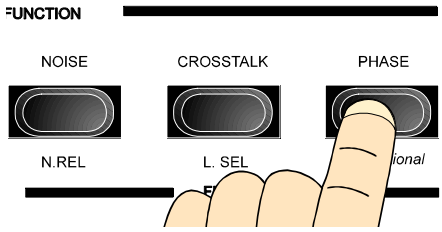


Fig. 4 Manual Activation of the Digital Analyzing Mode

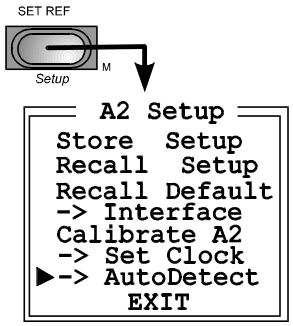
Automatic Activation

The digital board is equipped with a detection circuitry, that recognizes the signal type that is present at the XLR input. If a AES/EBU signal is detected, the digital analysis mode is enabled automatically and remains active, as long as the digital signal is present at the XLR input.

> ***NOTE The automatic signal detector monitors the XLR input connector at the front panel only, but not the SP-DIF and optical TOS link connectors.***

Disable Automatic Activation

The automatic domain detector can be disabled, so that the user has the full responsibility to set the appropriate analysis mode.
To disable the auto detect mode, double click to the <SET REF> key to enter the *A2 Setup* menu Therein, enter the submenu behind -> *AutoDetect* by using the <SCALE> softwheel. In the submenu, make your choice between *ON* / *OFF* and confirm by pressing the <SCALE> wheel once more.



Leaving the Digital Analysis Mode

If the automatic signal detector is disabled, the digital analysis mode can be left at any time by selecting an analog measurement function such as Level, THD+N, Noise etc.
Alternatively, i.e. as long as the auto detector is active, the operator has to make sure that no digital signal is present at the XLR input connector. The SPDIF and optical inputs do not feature an auto signal detector; their connection is not of importance.

> ***NOTE With the Auto-detector being active and a digital input signal being present at the XLR input, there is no way to leave the digital analysis mode.***

Display Overview

In the digital mode, the LC display will look similar to the one of ***Fig. 5***. Please notice, that the the layout of the screen may vary according to the actuial status of the unit. However, these variations have no influence on the following explanations.

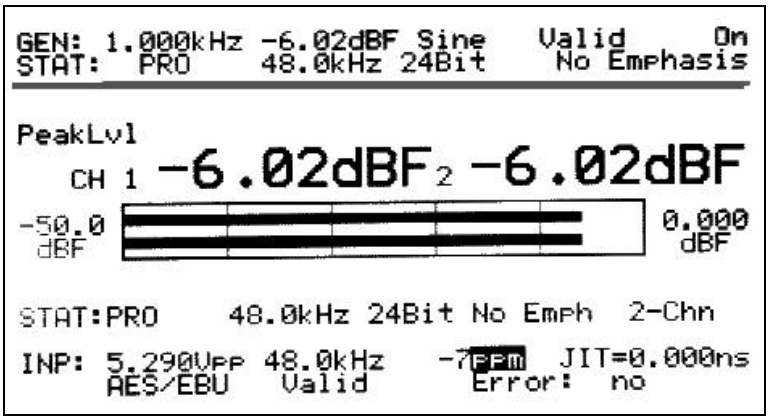

GEN: 1.000kHz -6.02dBF Sine Valid On
STAT: PRO 48.0kHz 24Bit No Emphasis
PeakLvl
CH 1 -6.02dBF 2 -6.02dBF
-50.0 dBF
0.000 dBF
STAT:PRO 48.0kHz 24Bit No Emph 2-Chn
INP: 5.290Vpp 48.0kHz -7PPm JIT=0.000ns
AES/EBU Valid Error: no

Fig. 5 Example of the Digital Display

Generator Settings

The top area of the screen is always reserved for the generator settings, regardless of the selected measurement function.

From left-to-right, the first line starts with the output frequency & level, followed by the output signal, the Validity status and the Channel status. The layout of the second line changes according to the selection of its leftmost entry.

- In the **STATus** mode, the signal format (**PRO**fessional or **Consum**er), the sampling frequency, the resolution and the Emphasis status are accessible.
- In the **DigOut** mode, the carrier level, frequency deviation as well as the jitter amplitude and modulation signal may be adjusted.
- In the **Config**uration mode, the output clock signal may be.
- In the **CRC** mode, the test sequence may be controlled in terms of its duration (hrs, min, sec) and transmission (start command).

More detailed explanations about these settings are filed in the chapter ***3Digital*** Generator.

Analyzer Results

The analyzer part takes all the space below the vertical line. It shows in its top left corner the name of the current measurement function, followed by the corresponding results, displayed in numerical figures and - depending on the measurement - also in form of an analog bar-graph.
The lower part of the display is reserved for the input signal status and parameters. Except of the **STATus** measurement mode, this part shows (from left-to-right and top-to-bottom) the acquired signal *Format*, *Sampling Frequency*, *Resolution*, *Emphasis* and *Channel Status*, *Carrier Level*, *Frequency Deviation*, *Jitter*, active *Input Connector* as well as the *Validity* and *Error* status. Please refer to ***7 AES/EBU Definitions*** for the specifications of AES/EBU signals.

More detailed explanations about these settings are filed in the chapter ***Digital Analyzer***.

Printouts

By pressing the <PRINT> button, the information of the actual display will be printed out to a connected printer.

```
NEUTRIK                                                    10.01.1996 10:33:18
------------------|--------|--------------------|--------|------------------
Output: Digital   |        | Input: Digital     |        |
SignalLvl:5.00Vpp | PRO    | SignalLvl:  4.98Vpp| PRO    | PeakLvl (Digital)
Sine              | 48.0kHz| Fs= 48000.24Hz     | 48.0kHz|
Valid             | No Emph| JIT=0.7ns    Valid | No Emph| 1:-2.00dBF
-2.00dBF 275.8 Hz |        | Error:       no    |        | 2:-2.00dBF
------------------|--------|--------------------|--------|------------------
```

Fig. 6 Printout of Digital METER Display

Mixed Signal Analysis

Apart from the purely analog or digital mode, the A2-D may be operated in the mixed signal mode as well. This means analog signal generation with digital analysis or digital signal generation with analog analysis.
The normal way to set-up the instrument in any of the available modes is to set first the analyzer, and then - if necessary - to modify the generator mode. This procedure is explained below.

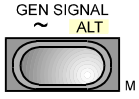
Digital Analysis / Analog Generation

Activate the digital analyzer as described in chapter ***Activation of the Digital Analyzing Mode*** on page ***10*** of this manual. Consequently, both the analyzer and the generator are set to the digital mode.
Now, there are two possibilities to set the generator to the analog mode.

- Place the cursor with the <SCALE> wheel to the Output Signal field in the top line of the LC display. Press the <SCALE> wheel and turn it until the keyword **Analog** appears. Confirm the selection by pressing the SCALE wheel. The generator status line changes then immediately to the known analog generation status information.

```
GEN: 1.000kHz -6.02dBF [Analog]  Valid      On
STAT:  PRO     48.0kHz 24Bit      No Emphasis
```

As long as the generator is in the digital analysis mode, the LED below the <GEN SIGNAL> button lights the *ALT* label. By pressing the <GEN SIGNAL> key once, the output signal toggles from *ALT* (digital output) to ~ (analog sine), i.e. the analog generator will be activated.



In this mode, the generator can be operated exactly the same way as in the analog only mode, except the following restrictions.

1. Only the sinusoidal output waveform may be generated.
2. When pressing the <IMPEDANCE> button, only the output impedance will change, but not the input impedance.
3. The level unit of the analog generator can be altered only, if the digital analyzer is set to the **PeakLvl** measurement function.

Application Example

A typical application of the A-D mixed mode is to measure the THD+N of an A/D converter vs. an increasing input level.

Fig. 7 shows such a measurement with and without level limiter. The limiter of the A/D converter increases the distortion already at a lower input levle, but decreases them in absolute terms. The other way round, i.e. without the limiter, the dynamic range of the A/D converter is higher, but the distortion increases to a higher end value at full scale.

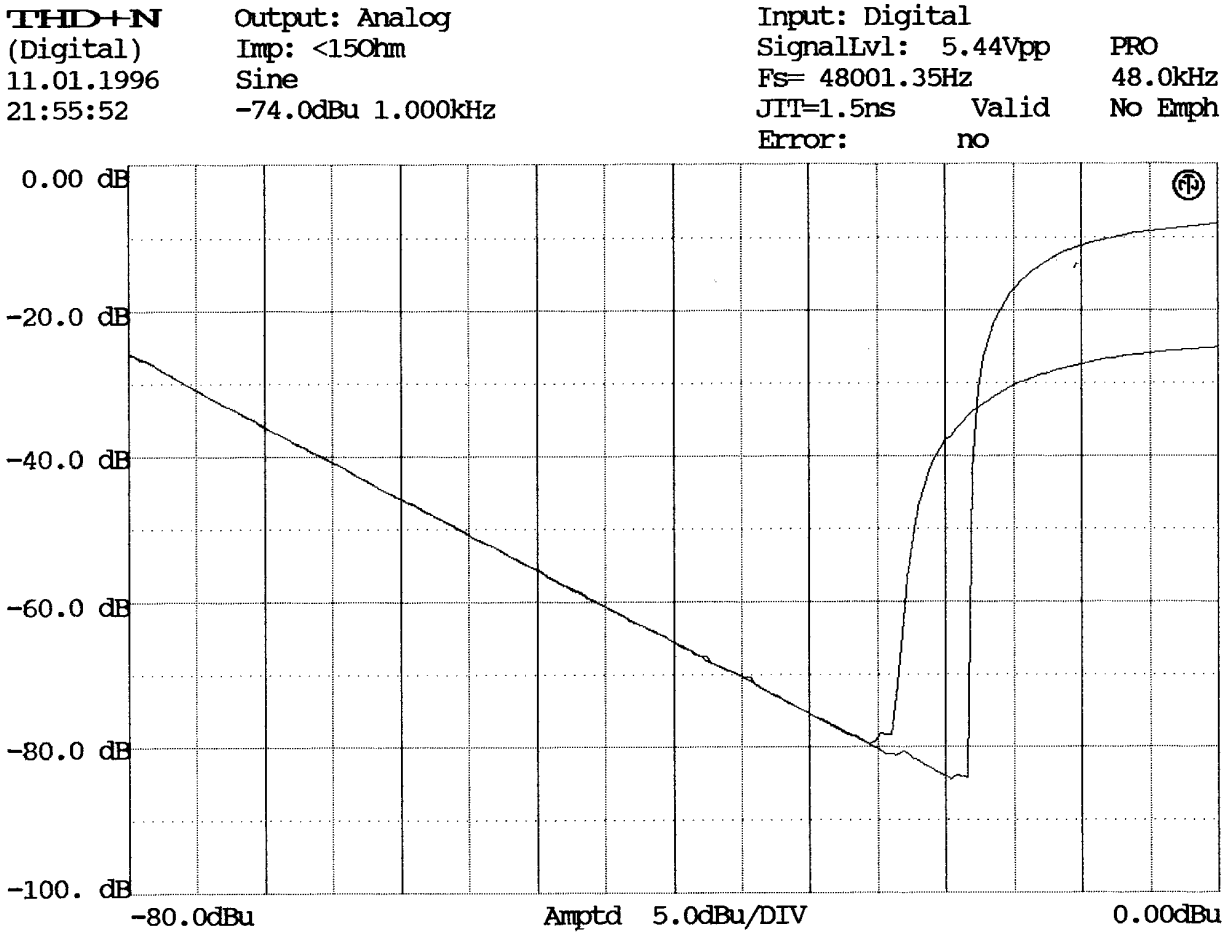


Fig. 7 Mixed Signal THD+N Response vs. Level

Switching Back to Digital Analysis

To get back to the analog generator mode, choose one of the two subsequent possibilities.

1. Press the <GEN SIGNAL> button to activate the *ALT* signal.
2. Place the cursor to the entry **Analog Sine** in the generator status line, and toggle to **Digital Sine** by using the <SCALE> softwheel.

Fig. 8 Generator Status Line in Analog Mode

Analog Analysis / Digital Generation

In order to switch the analyzer of the A2-D to the analog mode, while operating the generator in the digital mode, proceed as described in chapter ***Leaving the Digital Analysis Mode*** (p. ***10***).

As soon as the analog analysis mode is active, the digital output signal may be activated like any other alternate signal. Simply double-click to the <GEN SIGNAL> button, to open the *Alt. Signal* menu. Therein, either choose the *DigSine* or *DigSineDith* entry by using the <SCALE> wheel. Consequently, the generator will change into the digital mode, while the analyzer remains in the analog analysis mode.

In this mode, the digital generator is controlled through the three softwheels as explained in chapter ***3 Digital Generator***, while the analog analyzer may be operated through the keys on the A2-D front panel.

Operation Philosophy

Selections and Changes

In order to keep the user interface for the digital features as simple as possible, but without overloading the keyboard by triple functions or complex menus, all selections in the digital mode are performed through the <SCALE> softwheel only.

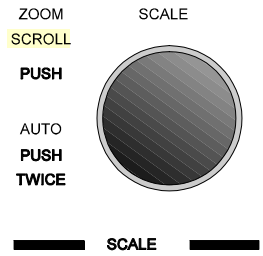


Fig. 9 <SCALE> Wheel

Each increment of the <SCALE> wheel moves the cursor (white on black) to the next editable entry. If one wants to alter the entry, simple press the wheel once, thus reverting the cursor to outline. This indicates, that the respective parameter may be changed. In this mode, an increment of the <SCALE> wheel alters the displayed entry to the next available one in the list.
To confirm a selection, press the <SCALE> wheel once, so that the cursor reverts back to inverse, while the selected entry becomes active.

Mode Switching

When switching between the modes of the A2-D (analog / digital), there are two rules to be considered.

1. Always change the mode of the analyzer first, and then - if required - the one of the generator.
2. When the analyzer mode is changed, the generator will always assume that status it had when the analyzer was in the respective mode the last time.

3 DIGITAL GENERATOR

The A2-D incorporates a fully digital generator, that allows to feed devices under test (DUTs) with all kinds of digital test signals. The information about the actaul settings of the generator, are indicated - like in the analog mode - on the top two lines of the LCD.

Fig. 10 Digital Generator Settings

The accessible entries of the first line are the output *Frequency* and *Peak Level* of the Audio signal, the *Output Signal*, the status of the *Validity Bit* and the *Output Status*.

The second line provides access to further parameters, that are used for the signal generation and the status coding. Its contents vary in accordance to the mode, that may be set through the leftmost entry of the line.

The following chapters describe in more details the accessible parameters of the generator.

Frequency

The frequency of the digitized audio signal may be set directly through the <FREQUENCY> softwheel. It may be set in the range from 10Hz up to half of the sampling frequency, which is 24kHz in maximum (at the standard clock rate of 48kHz).

NOTE ***With the audio signal frequency being set to exactly half the sampling frequency, the amplitude of the audio signal is not defined, due to the sampling theoreme.***

Peak Level

The *Peak Level* of the digitized audio signal may be varied directly through the <LEVEL> softwheel over the entire dynamic range of the selected resolution (max 24 bit).
The output level can be expressed in dBF (dB full scale) or in % (percent of full scale). To change the unit, the analyzer must be set to the measurement function **Peak Level**, wherein every press to the <UNIT> button toggles betwen the two available units. The calculation and relation of these two units is explained in more detail in chapter ***Units*** on page ***24***.

Output Signals

Sine (Digital)

The digital option offers different selections for the waveforms of the Audio signal output. The standard selection is the **Sine**, that corresponds to a (digitized) sinusoidal waveform with given frequency, level (amplitude) and resolution.

Pass

The second, basic audio signal mode is the **Pass** mode. It provides the same Audio signal at the generator output, that is actaully present at the analyzer input.

Nevertheless, the level and status of the output signal is created by the A2-D. In other terms, the digitized audio signal is routed through the unit, while its amplitude and status are accessible through the A2-D.
In this way, the operator may change all status information (e.g. resetting the COPY protection bit), the signal Format (PRO or Consum), the Emphasis, the word Resolution, the audio Output Level, the Carrier Level etc. according to the demands of the application.
Additionally, the Pass mode allows to use the A2-D as an interface between two digital devices. In this case, the sampling frequency is automatically locked to the one at the analyzer input, indicated by the keyword **InpClk** on the second line of the LC display. Every attempt to change this setting in the Pass mode creates an error message.

Not available in Pass Mode!

Sine+D / Pass+D (Dithering)

In both the **Sine** and the **Pass** mode, dithering may be added to the generated signal. Dithering is a noise signal, with defined statistical characteristics, that is added to the LSB of a signal, in order to improve the conversion quality of a reduced resolution.

If, for instance, a 20bit sine signal with a low level is recorded on a 16bit DAT recorder, it will be truncated to the 16bit resolution. Unfortunately, this process changes the waveform of the original low-level sine signal to almost square shape, which causes high distortions.
However, by dithering, i.e. adding 'digital noise' to the LSB of the original sine signal, the waveform of the truncated signal will remain much closer to the one of the original signal. The price for the thus reduced distortions at low levels is an overall increase of the residual noise floor of about 5dB (see ***Fig. 12***).

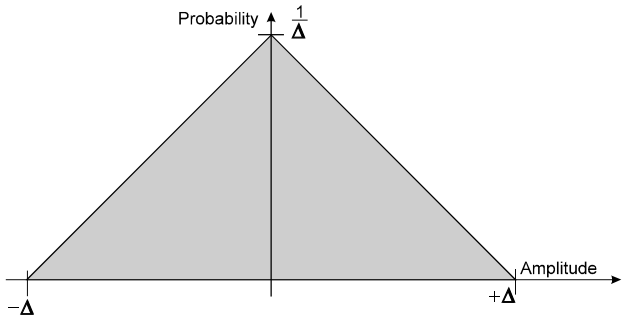


Fig. 11 TPDF Dither

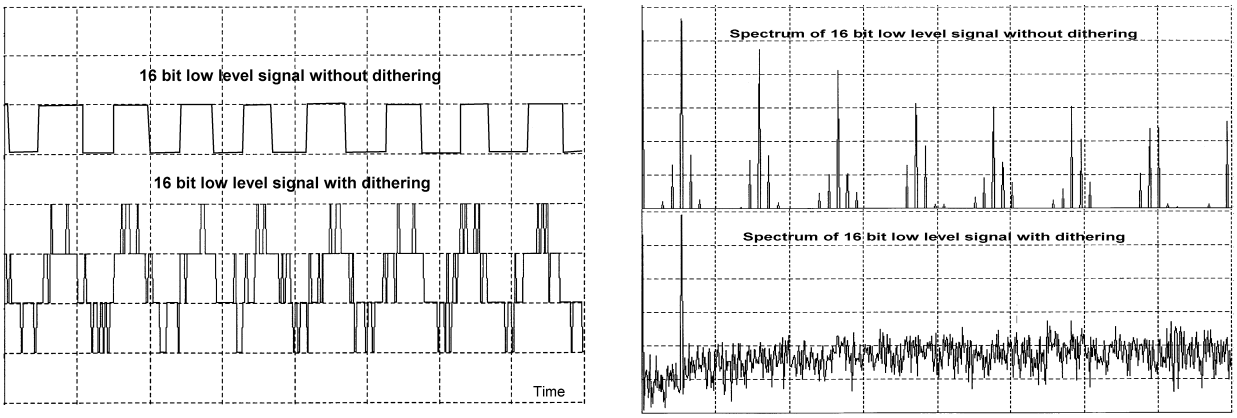

16 bit low level signal without dithering
16 bit low level signal with dithering
Time
Spectrum of 16 bit low level signal without dithering
Spectrum of 16 bit low level signal with dithering

Fig. 12 Dithering

The implemented dither algorithm of the A2-D is the TPDF dither. This is a noise signal with **T**riangle **P**robability **D**ensity **F**unction. It is a white noise signal with a triangular probability distribution and an amplitude of ±1 LSB (see ***Fig. 11***).

CRC

Basically, the CRC test of the A2-D allows to check, whether a purely digital signal transmission provides bit errors or not. This is achieved by sending a noise-like bit sequence through the signal path, and comparing the CRC checksums of the sended and the received sequence against each other. If both results are equal, no bit errors have occured.

Normally, the CRC test is executed through a single A2-D unit, acting both as generator of the CRC test sequence and its analyzer. This mode is described in full details in chapter ***CRC Test***. However, it is also possible to create a CRC test sequence on one unit, and analyzing it on another one. For this purpose, the output signal may be set to **CRC**, without the need to switch the analyzer into the corresponding mode.

If the **CRC** selection has been activated, the second line of the LC display will automatically show the corresponding entries. Please refer to chapter ***CRC Line*** for further details.

Analog (Sine)

The last entry of the output signal selection is labeled with **Analog**. When activating this selection, the A2-D switches to analog signal generation, while the analyzer remains in the digital mode.
To switch back to digital signal generation, enter the *Alt. Signal* menu behind the <GEN SIGNAL> button.

NOTE Alternatively, The analog generator mode can be activated by setting the <GEN SIGNAL> button to „~“ instead of ALT.

Validity Bit

As it name says, the Validity bit indicates, whether the audio signal is valid or not (see chapter ***7 AES/EBU Definitions***). It can be toggled between **Valid** and **InValid** in any digital signal generation mode.

Output Status

The output status entry filed is located on the right hand side of the top line in the LC display. It allows to define, whether the two audio channels, that are encoded in an AES-3 signal, shall be active or muted.

The four available states are

- **On**: both channels are active
- **1 On** / **2 On**: the indicated channel is active only, the other one is muted
- **Mute**: both channels are muted.

NOTE Do not mismatch the Output Status with the physical output connectors. The AES-3 signal will always be present at output connector OUT A only, even when both audio channels are muted.

STATus Line

The **STAT**us line may be displayed by entering the leftmost entry of the second generator line. It provides access to the *Signal Format*, the *Sampling Frequency*, the word *Resolution*, the *Emphasis* and the *Channel Status*.

Signal Format

The *Signal Format* is either **PRO**fessional or **Consum**er (see ***7 AES/EBU Definitions***). The corresponding status may be monitored best by entering the **Status** measurement function.

Sampling Frequency

The third field in the **STAT**us mode line allows to set the *Sampling Frequency* of the generator to one of the following values.

- **32.0kHz** (standard AES-3 sampling frequency)
- **44.1kHz** (standard AES-3 sampling frequency)
- **48.0kHz** (standard AES-3 sampling frequency)
- **Inp Clk** the generator synchronizes itself onto the clock of the input signal
- **fs-Sync** the sampling frequency generator is synchronized by the external clock, that must be fed to the *fs SYNC INPUT* on the rear panel.

Resolution

The *Word Resolution* indicates the number of bits to quantizise the audio signal. It may be set to one of the following values: **4Bit**, **8Bit**, **10Bit**, **12Bit**, **16Bit**, **18Bit**, **20Bit**, **22Bit**, **24Bit**.

Emphasis

This field allows to activate a frequency response correction in order to improve the dynamic range of older digital recordings.

Following selections are available.

Emphasis Filter Type	**PRO**fessional Mode	**Consum**er Mode
NoEmphasis	✓	✓
50/15µ**s**	✓	✓
CCITT	✓	-

Table 1 Emphasis Filter Availability

Channel Status

The last entry of the **STAT**us mode line refers to the *Channel Status* of the audio signal.

- **Stereo** indicates, that the digitized audio signal is a true stereo signal.
- **2-Chn** indicates, that the two channels of the digitized audio signal are completely independent, i.e. not a stereo signal.
- **1-Chn** indicates, that both audio signal channels transmit the identical 1-channel signal.

DigOut Line

The **DigOut** line can be activated by placing the cursor with the <SCALE> softwheel to the leftmost field in the second line (from top of the LCD), and pressing the wheel to enter the selection. Increment the wheel, until the setting **DigOut** appears, and confirm by pressing the <SCALE> wheel once more.

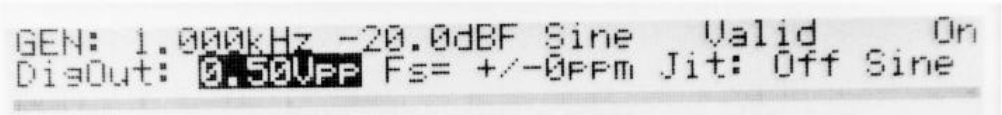


Fig. 13 ***DigOut*** *Line*

> ***NOTE*** ***All changes in the DigOut line are active only, if this menu is open. They will be reset to default values as soon as the menu is left. This ensures, that in normal operation the DUT is fed with a proper signal, according to the AES3 standard.***

Carrier Output Voltage

The first accessible parameter of the **DigOut** line shows a number, labeled with **Vpp**. It indicates the *Carrier Output Voltage*, that normally equals 5.0Vpp.

In order to simulate long cables, or to test the safety margin of the device under test (DUT), the A2-D offers the possibility to reduce the Carrier Output Level in 12 steps from 5.0Vpp down to 0.15Vpp for the **PRO**fessional format, and from 0.5Vpp down to 0.015Vpp to for the **Consum**er format.

> ***NOTE*** ***In the Consumer mode, the Carrier Level value is scaled by factor 10. This means, that a reading of 4.0Vpp actually corresponds to an amplitude of 0.4Vpp.***

Fs De-tuning

In order to test the locking range of a device under test (DUT), the *Sampling Frequency* of the generator may be de-tuned from the standard values in the range of ±1500ppm.
Keep in mind, that this mode is only active, if the digital generator is locked to one of the three standardized sampling frequencies (32.0kHz, 44.1kHz, 48kHz). In the **Inp Clk** and **fs-Sync** mode (see ***STATus Line***), no frequency de-tuning is possible.

> ***NOTE*** ***As long as the Fs De-tuning is set to a value [1] ±0ppm, no Jitter can be added to the output signal, since both features use the same internal circuitry.***

Jitter Amplitude

Jitter is defined as the phase (i.e. time) variations between the ideal and the actual transients of a digital bit stream. In other terms, due to poor clock circuitry or die to long cables, it may happen that small variations of the ideal sampling-instants of an A/D converter occur. These variations are called *Jitter*.
The influence of jitter becomes best noticeable when performing a THD+N analysis of the modified signal, since jitter creates a non-linearity in the transfer curve, resulting in increased distortion values.

The A2-D features a special mode, where a user-defined **Jit**ter value may be added to the output signal. This powerful tool allows to test, how sensitive a DUT reacts to a certain amount of jitter.
The **Jit**ter value itself has to be understood as the total jitter amplitude. For instance, a value of 20ns means, that (peak) time variations of ±10ns from the ideal sampling instants are produced.

Jitter Modulation

Right of the **Jit**ter amplitude, the last field in the **DigOut** line provides access to the *Modulation Signal* of the jitter.
To understand the meaning of this signal, be aware that the variations explained above, don't necessarily alsways have the same amplitude, but actually are distributed over the selected range. Consequently, a modulation signal must make sure, that this distribution follows a certain pattern.

In the A2-D, there are two modulation signals implemented, **Sine** and **Noise**.

- In the **Sine** mode, the jitter modulator is a sinusoidal signal with a fixed frequency of 2.5kHz.
- In the **Noise** mode, the jitter modulator is a white noise source, producing an equal distribution of the jitter. The bandwidth of the white noise signal is 20Hz to 20kHz.

NOTE** **As long as the generator adds Jitter to the output signal, the sampling frequency cannot be de-tuned, since both features use the same internal circuitry.

Configuration Line

The A2-D provides a BNC trigger output at the rear panel, labeled *X/Y/Z TRIG. OUTPUT*. The type of clock, that is fed to this connector, can be configured in the **Config**uration line by the user.
To enter the **Config**uration line, place the cursor with the <SCALE> softwheel to the leftmost field in the second line (from the top of the LCD), and press the wheel to open the selection. Increment the wheel, until the setting **Config** appears, and confirm by pressing the <SCALE> wheel once more.

Four different clocks are available.

Name	*Description*
TrigX	Generates a trigger signal for each occurrence of the X-preample. (Channel 1 word start)
TrigY	Generates a trigger signal for each occurrence of the Y-preample. (Channel 2 word start)
TrigZ	Generates a trigger signal for each occurrence of the Z-preample. (Channel 1 word start at the beginning of a new block)
EyePatt	Generates the bit clock of the data stream. The bit clock is 64 times the sampling frequency, (2*32 bit)

Table 2 Trigger Output Signals

NOTE The trigger output is generated by the analyzer, and not by the generator.

CRC Line

This line provides access to the output parameter of a CRC test signal, i.e. the length of the sent-out sequence, and allows the operator to start the transmission.
To enter the **CRC** line, place the cursor with the <SCALE> softwheel to the leftmost field in the second line (from the top of the LCD), and press the wheel to open the selection. Increment the wheel, until the setting **CRC** appears, and confirm by pressing the <SCALE> wheel once more.

Fo further details on the CRC test, please refer to chapter ***CRC Test***.

Start

By confirming this entry, the transmission of the CRC test sequence will be started immediately. Please notice, that the duration of the sequence has to be set first (see below).

h, m, s (Sequence Duration)

These three entry fields labeled h, m, s allow to define the duration of the CRC test sequence in hours, minutes and seconds.

4 DIGITAL ANALYZER

In the digital mode, The A2-D offers a broad variety of measurements results - some of them are displayed continuously, others to be activated by the user. Most of these measurement functions are gained in the digital domain only, i.e. without conversion into the analog domain (see next chapter).
However, some of the measurement results are obtained after the D/A conversion of the digitized audio data into an analog signal. They are explained in chapter ***Analog Measurement Functions***.

Digital Measurement Functions

To select the main measurement function, place the cursor to the top left entry of the analyzer display (below the horizontal line) by turning the <SCALE> softwheel.

Press the wheel once to enter the selection mode, indicated by the outlined cursor. Each increment of the <SCALE> wheel switches to another measurement function.

In this way, choose the required function and press the <SCALE> wheel once to confirm and activate the selection.

Peak Level

The **Peak Lvl** function calculates the actaul peak value out of the incoming digital audio signal.

The result is displayed for both channels simultaneously in numerical letters and in analog form as a bar-graph. Please notice, that the reading of the bar-graph is much faster than the numerical display, thus allowing to monitor fast trends and tendencies.

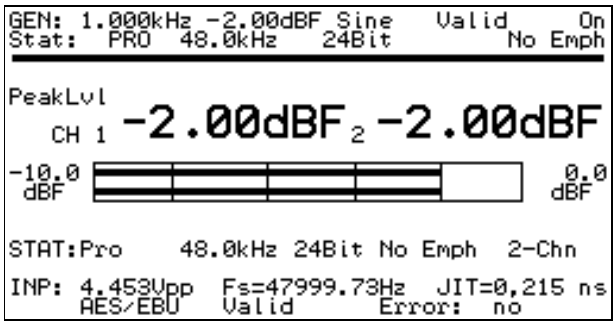

GEN: 1.000kHz -2.00dBF Sine Valid On
Stat: PRO 48.0kHz 24Bit No Emph
PeakLvl
CH 1 -2.00dBF 2 -2.00dBF
-10.0 dBF
0.0 dBF
STAT:Pro 48.0kHz 24Bit No Emph 2-Chn
INP: 4.453Vpp Fs=47999.73Hz JIT=0,215 ns
AES/EBU Valid Error: no

Fig. 14 Peak Level Screen

Printout

To print out the contents of the **Peak Lvl** screen, connect a printer to the Centronics connector on the rear panel of the unit and press the <PRINT> button. ***Fig. 15*** shows an example of such a printout.

NEUTRIK 10.01.1996 10:33:18

Output: Digital		Input: Digital		
SignalLvl:5.00Vpp	PRO	SignalLvl: 4.98Vpp	PRO	PeakLvl (Digital)
Sine	48.0kHz	Fs= 48000.24Hz	48.0kHz	
Valid	No Emph	JIT=0.7ns Valid	No Emph	1:−2.00dBF
-2.00dBF 275.8 Hz		Error: no		2:−2.00dBF

Fig. 15 Printout of Peak Level Measurement

Units

The peak level can be displayed in two different units, dBF (decibel full scale) and % (percent of full scale). Obviously, the peak level units refer to the maximum possible level of the digital signal (100% or 0 dBF).

Unit	*Type*	*Reference*	*Calculation*
%	Linear	Full scale (100%)	$\frac{Actual\_Value}{Max\_Value}$
dBF	Logarithmic	Full scale (0dBF)	$20 * \log_{10}(\frac{Actual\_Value}{Max\_Value})$

Table 3 Digital Peak Level Units

Status Display

The **Status** of the digital audio signal is encoded in the subframes of the AES-3 bit stream. Every 192 frames, the status is repeated (see chapter *7 AES/EBU Definitions*).

To activate the **Status** measurement function, place the cursor to the top left field of the analyzer screen. Press the <SCALE> wheel once and turn it until **Status** appears. Press the <SCALE> wheel once more to confirm.

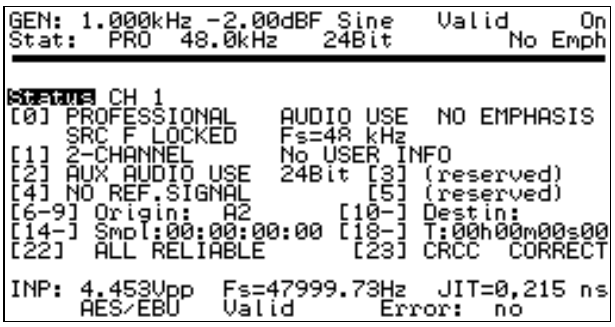

GEN: 1.000kHz -2.00dBF Sine Valid On
Stat: PRO 48.0kHz 24Bit No Emph
Status CH 1
[0] PROFESSIONAL AUDIO USE NO EMPHASIS
SRC F LOCKED Fs=48 kHz
[1] 2-CHANNEL No USER INFO
[2] AUX AUDIO USE 24Bit [3] (reserved)
[4] NO REF.SIGNAL [5] (reserved)
[6-9] Origin: A2 [10-] Destin:
[14-] Smpl:00:00:00:00 [18-] T:00h00m00s00
[22] ALL RELIABLE [23] CRCC CORRECT
INP: 4.453Vpp Fs=47999.73Hz JIT=0,215 ns
AES/EBU Valid Error: no

Fig. 16 Status Screen (Professional Format)

The A2-D directly translates the contents of the status bits and displays the result on the **Status** screen, thus enabling the user to directly read the meanings.
The proper interpretation of the status bits for both the Professional and the Consumer format is done automatically by the A2-D. Therefore, there are two different layouts of the **Status** screen.

Printout

To print out the contents of the **Status** screen, connect a printer to the Centronics connector on the rear panel of the unit and press the <PRINT> button. ***Fig. 16*** shows an example of such a printout.

```
NEUTRIK                                                    10.01.1996 10:38:42
------------------|---------|-------------------------------------------------
Output: Digital   |         | Input: Digital
SignalLvl:5.00Vpp | PRO     | Status CH 1
Sine              | 48.0kHz |       [0] PROFESSIONAL   AUDIO USE  NO EMPHASIS
Valid             | No Emph |           SRC F LOCKED   Fs=48 kHz
-2.00dBF 10.80 Hz |         |       [1] STEREO MOD     NO USER INFO
                  |         |       [2] AUX AUDIO USE  24Bit [3] (reserved)
                  |         |       [4] NO REF.SIGNAL        [5] (reserved)
                  |         |       [6-9] Origin:  A2      [10-] Destin:
                  |         |       [14-] Smpl:00:00:00:00 [18-] T:00h00m00s00
                  |         |       [22] Byte:ALL RELIABLE  [23] CRCC  CORRECT
------------------|---------|-------------------------------------------------
```

Fig. 17 Printout of Status Screen (Professional Format)

Bit Statistics

Another measurement function of big help for the test and diagnosis of a system is the **BitStat**istics function. It visualizes the state of all bits in the digital audio signal, covering the *AUX* bits, the audio data as well as the *User*, *Validity*, *Status* & *Parity* bits.

The **BitStat** screen allows to see quickly, which bits of the audio data are permanently low (0), high (1) or changing (⇳).

In this way, e.g. the resolution of the encoded audio word can be verified immediately. Just check, how many of the 24 bits - below the AUX and MSB labels - are actually changing, indicated by the ⇳ sign.
If some of the bits are constantly 0, the resolution of the audio signal is obviously less than the maximum of 24bits.

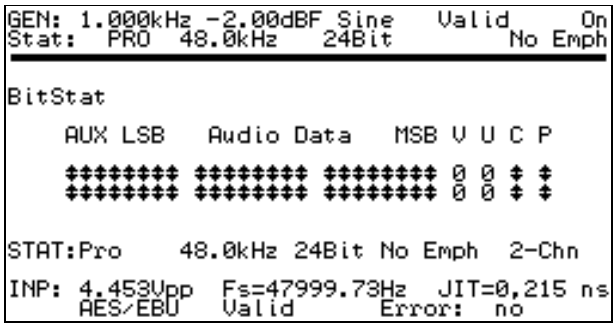

GEN: 1.000kHz -2.00dBF Sine Valid On
Stat: PRO 48.0kHz 24Bit No Emph
BitStat
AUX LSB Audio Data MSB V U C P
STAT:Pro 48.0kHz 24Bit No Emph 2-Chn
INP: 4.453Vpp Fs=47999.73Hz JIT=0,215 ns
AES/EBU Valid Error: no

Fig. 18 Bit Statistic Screen

If the *Validity* bit is permanently 0, the incoming data are valid. The *User* bit is not used, i.e. constantly 0. The *Status* and the *Parity* bit are both normally changing.

Printout

To print out the contents of the **Status** screen, connect a printer to the Centronics connector on the rear panel of the unit and press the <PRINT> button. ***Fig. 16*** shows an example of such a printout.

```
NEUTRIK                                                    10.01.1996 10:39:47
-----------------|---------|-------------------------------------------------
Output: Digital  |         | Input: Digital
SignalLvl:5.00Vpp| PRO     | BitStatistic
Sine         1 On| 44.1kHz |                AUX LSB   Audio Data   MSB V U C P
Valid            | No Emph |            1: 000000XX XXXXXXXX XXXXXXXX 0 0 X X
-2.00dBF 10.80 Hz|         |            2: 00000000 00000000 00000000 0 0 X X
-----------------|---------|-------------------------------------------------
```

Fig. 19 Printout of Bit Statistic Screen

CRC Test

The **CRC Test** allows to check the transmission reliability of a digital DUT. For this purpose, the A2-D generates a defined pseudo-random noise signal, i.e. a reproducible data sequence, which is transmitted through the DUT.
Next, the analyzer calculates a CRC (Cyclic Redundance Check) Hex-word over the received data sequence and displays it to the user.
Finally, the whole test is repeated with the same data sequence in order to compare the results. If an identical CRC Hex-word results, the DUT has transmitted the sequence without any bit losses or changes. However, in case that the CRC Hex-word changes between two or more transmissions, the DUT obviously modifies the incoming bit stream in an unpredictable manner.

NOTE ***Be aware that any change in gain, frequency, jitter etc. changes the CRC result. The CRC test is intended to compare whether digital DUTs are completely identical (e.g. in production)***

To define the CRC test settings, proceed as follows by using the <SCALE> softwheel.

- Set the generator signal to **CRC**.
- In the second line, enter the desired data sequence duration (hours, min, seconds).
- Set the analyzer mode to **CRC Test**.
- Enter the desired analyzing duration (h, m, s).
- Start the CRC test through the filed **Start** in the second generator line.

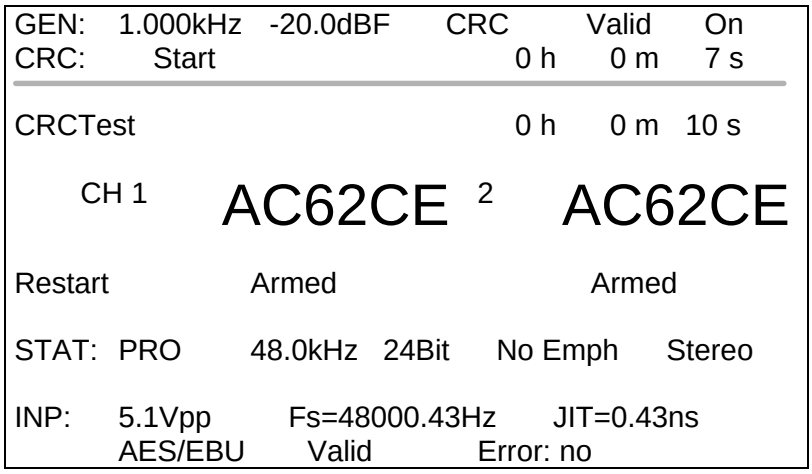


Fig. 20 CRC Test Screen

- As long as a signal is present at the input, the analyzer status (right of the **Restart** field) shows **Waiting**.
- To start a CRC test, at least one second of audio silence (zero input) must be present. During this audio silence (on one or both channels), the status indicator switches to **Audio Silence**. After one second of audio silence, the analyzer shows **Armed**.
- As soon as a non-zero audio word is received, the respective channel switches to **Running** while the result shows **-Busy-** in large characters until the measurement has been finished.

During the test, you can watch the progress on the CRC test progress in form of a countdown for both the generator and analyzer test duration. After expiration of the defined analyzing time, the 6-digit CRC Hex-result is displayed.

The independently definable generator and analyzer settings allow extended testing possibilities of digital DUTs.

- For instance, the longer the generator and analyzing duration are, the larger becomes the variety of transmitted words within the data sequence, i.e. the tougher becomes the test.
- A generator duration far longer than the analyzing one allows to measure other audio data after completed CRC test.
- Changes of other setup settings such as frequency, level, word length etc. allow to test digital DUTs under a wide range of conditions.

NOTE ***Please notice, that for direct comparison between two or more measurement results, no changes of the generator and analyzer settings are allowed.***

Analog Measurement Functions

The digital interface board AO10 is equipped with a high performance D/A converter, that allows to link the received digital signal to the analog analyzer of the A2. In this way, analog measurements of the incoming AES/EBU signal may be performed as well.

RMS Level

The digital Peak Level measurement, described previously, provides information about the peak-to-peak signal level, compared to the full scale of the converter.

On the other hand, with the analog **RmsLvl** measurement function, it is also possible to measure the RMS level of the digital input signal. As reference level, a sine signal with 0dBF peak-to-peak level is assumed.

To activate the RMS Level measurement function, place the cursor to the top left field on the analyzer screen, press to the <SCALE> wheel to open the selection and choose **RmsLvl**. Confirm by pressing the wheel once more.

Since this is a relative unit, measured against a fix reference (i.e. full scale), the available units are %, *1, and dBr.

THD+N

To activate the **THD+N** measurement function, place the cursor to the top left field on the analyzer screen, press to the <SCALE> wheel to open the selection and choose **THD+N**. Confirm by pressing the wheel once more.

The **THD+N** analysis may be very useful for the analysis of single tone test signals (sine). It allows quickly to verify the quality of the digital transmission. A 16bit audio transmission should achieve a THD+N value in the range of -90dB. With reduced data width or artificially enlarged word width, the THD value will even stay below the theoretical value. Also the influence of the jitter to the THD+N value is notable when performing analog distortion measurements.

Monitor Speaker / Headphone Output

The D/A converter always links the incoming audio signal to the analog analyzer. Therefore, it is possible to listen to the incoming audio signal at any time.

The signal is fed to the internal loudspeaker, to the BNC output of channel A (normalized to 1V) as well as to the headphone output at the rear panel.

Analyzer STATus Line

Every AES/EBU signal incorporates not only the audio data, but also some status information and codes that indictae the validity of the sample.
The **STAT**us line of the analyzer display shows some status parameters of the input signal.

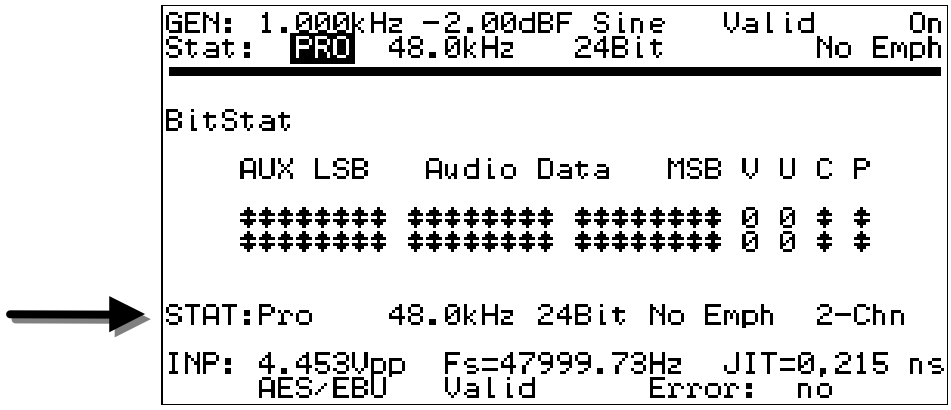

GEN: 1.000kHz -2.00dBF Sine Valid On
Stat: PRO 48.0kHz 24Bit No Emph
BitStat
AUX LSB Audio Data MSB V U C P
STAT:Pro 48.0kHz 24Bit No Emph 2-Chn
INP: 4.453Vpp Fs=47999.73Hz JIT=0,215 ns
AES/EBU Valid Error: no

Fig. 21 Digital Display with Analyzer Status Line

Although this information does not fully describe the signal status, it reflects at least the most important values.

PROfessional or **Consum**er indicates, whether the received signal has professional or consumer format. Depending on this format, the remaining states differ from each other (see also chapter ***7 AES/EBU Definitions***).

Regardless of the signal format, the next entry of the analyzer **STAT**us line refelcts the clock of the input signal. It shows the sampling rate, with which the audio data have been digitized. Standard values are: 32.0kHz, 44.1kHz and 48.0kHz. However, in the **InpClk** mode, the numerical result disappears, so that the input clock can be obtained via the **Fs** field only (see below).

STATus Line in PROfessional Format

The leftmost entry of the line indicates the signal format (**PRO**fessional), followed by the *Input Clock*. Please notice, that the displayed value always equals one of the 3 standard sampling rates (32.0kHz, 44.1kHz or 48.0kHz).

Next, the *Resolution* indicates with how many bits per sample the Audio data have been coded. Maximum value is 24Bits.

Follows the *Emphasis* field, indicating whether an emphasis filter type has been applied on the audio data or not. Refer also to chapter ***Emphasis*** on page ***18***.

The last statement in the line covers the *Channel Status*, indicating what kind of audio material is transmitted (**Stereo**, **2-Chn** or **1-Chn**).

STATus Line in Consumer Format

The leftmost entry of the line indicates the signal format (**Consum**er), followed by the *Input Clock*. Please notice, that the displayed value always equals one of the 3 standard sampling rates (32.0kHz, 44.1kHz or 48.0kHz).

Next, the *Category Code* indicates whether the source of the received audio data is of type **General**, **CD**, **DAT** or **Exprmtl**.

Follows the *Emphasis* field, indicating whether an emphasis filter type has been applied on the audio data or not. Refer also to chapter ***Emphasis*** on page ***18***.

The last statement in the line covers the *Copy Status*, indicating whether the incoming audio data are copy protected (**NoCopy**) or not (**CopyOk**).

Analyzer INPut Lines

On the bottom of the digital LC display, two **INP**ut lines provide additional information about the incoming signal. In contrary to the analyzer **STAT**us line, the content of the **INP**ut lines are not depending of the input signal format.

Carrier Level

The leftmost reading in the upper analyzer **INP**ut line continuously shows the *Carrier Level*, expressed in **Vpp**, regardless of the selected measurement function. It measures the physical peak-to-peak amplitude of the input signal, with a bandwidth-limiter applied to filter-out possible transients of the slopes.

The *Carrier Level* may help to detect level, impedance matching and cable-loss problems.
Please notice, that the specification for AES/EBU signals allows carrier levels in the range of 200mVpp to 5Vpp while the IEC958 format specifies levels in the range of 200mVpp to 500mVpp only.

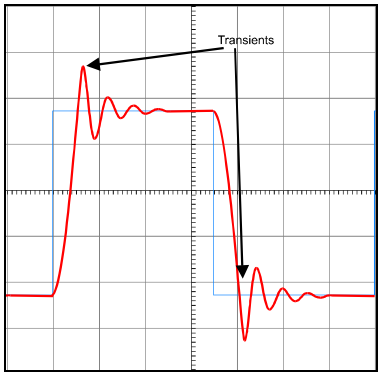


Fig. 22 Carrier Level with Ringing

Sampling Frequency

The actaul sampling frequency **Fs** of the input signal is continuously monitored and displayed in all measurement functions.

Actually, there are two display modes to show the sampling frequency.

- As absolute value, expressed in Hz (e.g. 48000.12Hz)
- As standard frequency ± a relative frequency deviation, expressed in ppm (e.g. 48.0 kHz -23ppm).

Fig. 23 Relative Sampling Frequency Display

To change from the absolute to the relative sampling frequency display, place the cursor with the <SCALE> wheel to the **Hz** label behind the sampling frequency. Select this item by pressing the wheel once and toggle to **ppm**. Press the <SCALE> wheel once more to confirm.

Jitter

The A2-D continuously monitors the peak-to-peak *Word Clock Jitter* of the incoming AES-3 signal. The measured jitter value is displayed in the field behind the keyword **JIT.** It is expressed in **ns** (1 nanosecond = $^{1}/_{1000}$ of a microsecond).

Alternatively to the absolute **ns** unit, jitter can also be expressed in a relative unit, the Unit interval **UI**. **UI** expresses the amount of jitter compared to the duration of a frame, thus being dependent of the sampling frequency. The conversion from the absolute **ns** to the relative **UI** unit and vice versa can be done according to the subsequent formulas.

$$dBUI = 20 * \log_{10}\left(\frac{Jitter[s]}{1/f_s}\right) = 20 * \log_{10}\left(Jitter[ns] * 10^{-9} * f_s[Hz]\right)$$

Equation 1 Conversion from ns to UI

$$Jitter[s] = 10^{(UI * \frac{1}{f_s[Hz]})} \quad or \quad Jitter[ns] = 10^{(\frac{UI*10^9}{f_s[Hz]})}$$

Equation 2 Conversion from UI to ns

Fig. 24 shows how jitter looks like if displayed on an oscilloscope. Keep in mind, that due to the high speed of the time deviations, the bandwidth of such a scope has be very high, i.e. in the range of 200MHz.

Since the A2-D is an Audio measurement system, the bandwidth of the jitter measurement is set from 700Hz to 20kHz to cover the full audio band up to the upper edge. The low frequency limit is set to avoid measurements of the sampling frequency drift and deviation.

The AES standard specifies, that the jitter value must be below ±20ns for proper operation. The A2-D is capable to measure jitter values from the specified maximum down to values as low as low as ~0.5ns.

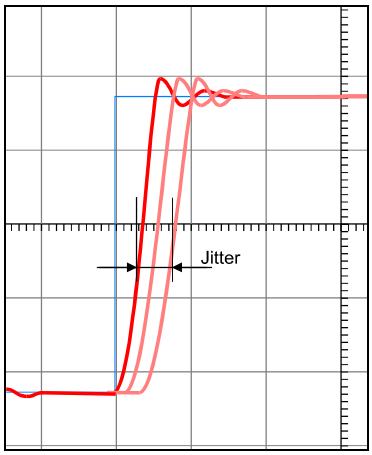


Fig. 24 Jitter

Input Connector

The leftmost entry on the second analyzer **INP**ut line allows the operator to select the input connector for the signal to be analyzed.

To activate the required connector, place the cursor with the <SCALE> softwheel to this field, push it once and select either **AES/EBU** (XLR connector CH A on front panel), **SPDIF** (RCA connector on rear panel) or **Optical** (TOS link on rear panel).

Validity Bit

The *Validity* information of the input signal is displayed on the bottom line of the screen. The keyword **Valid** indicates, that the received audio material is marked as valid, while in case of invalid samples the status changes to **Invalid**.

Errors

The last reading of the bottom line indicates the *Errors* that occurred during the signal transmission.

Error State	***Description***
No	No error has been detected, the received digital information is correct.
Confidence	Problems to properly read the incoming signal, e.g. due to high jitter or a too low carrier level. The audio information may be faulty.
Code	Problems in the biphase coding of the signal - unreliable audio information.
Parity	Parity errors, caused by one or more bits, that have changed during the transmission. Nether status nor audio data must be used.
Unlock	The analyzer is unable to read and interpret the incoming signal as a digital bit stream, so that all further digital analysis is useless.

Table 4 Error States

5 DIGITAL SWEEPS

As in the analog mode, the A2-D may also perform sweeps on digital signals. To do this, simply select the digital measurement function in the METER display mode and switch to the GRAPH display mode. Enter the setup with start value, stop value, resolution, scaling etc., and press the <DO SWEEP> button to let the digital sweep be recorded automatically. To store and recall recorded graphs, apply the same procedure as for analog sweeps.

> ***NOTE*** ***In opposition to the analog mode, the digital measurement function can be changed in the METER display mode only.***
>
> ***If the AutoDetect mode is active, and a digital signal is applied to the AES3 inputs, the analog measurement function buttons are inactive.***
> ***If the AutoDetect mode is switched off, any press to a measurement function button switches to the corresponding analog analysis mode.***

Following sweep modes are supported in the digital domain.

- Frequency sweeps from 10Hz to fs/2 with constant level.
- Tape frequency sweeps from 10Hz to fs/2 with fixed duration per step.
- Amplitude sweeps from -120dBF to 0dBF at a fixed frequency.
- Time sweeps with a step interval from 3s up to 180hours.
- Table sweeps with freely definable frequency/amplitude pairs.

Like in the analog mode, digital sweeps may be synchronized internally or externally.

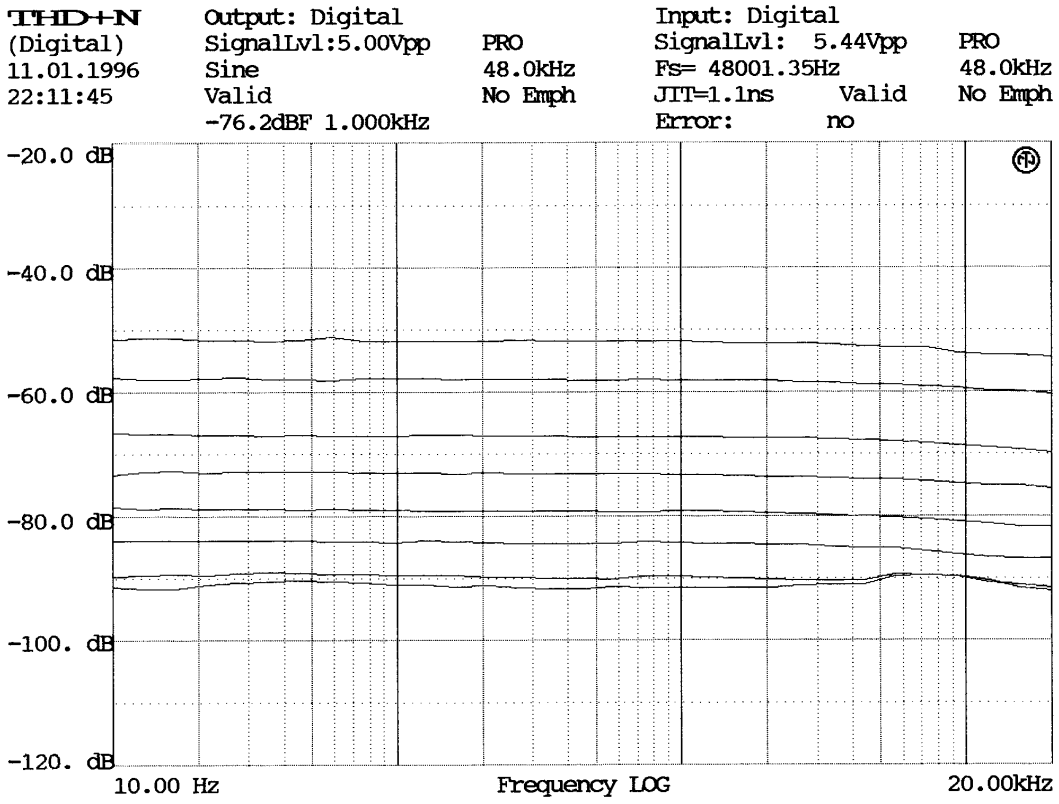


Fig. 25 Digital THD+N Sweep of a DAT Recorder at Various Levels

6 FFT ANALYSIS

Basically, the digital option AO10 was designed to generate & analyze digital signals according to the AES-3 and SPDIF standards. However, due to the computing power of the involved DSP, the A2-D can in principal also calculate an FFT (Fast Fourier Transformation) spektrum analysis on an analog input signal.

The FFT option (article code AO12) adds an 18 bit A/D converter module and updates the firmware with the DSP code do to the FFT calculations. The modified block diagram of the AO10 with the option AO12 is shown below.

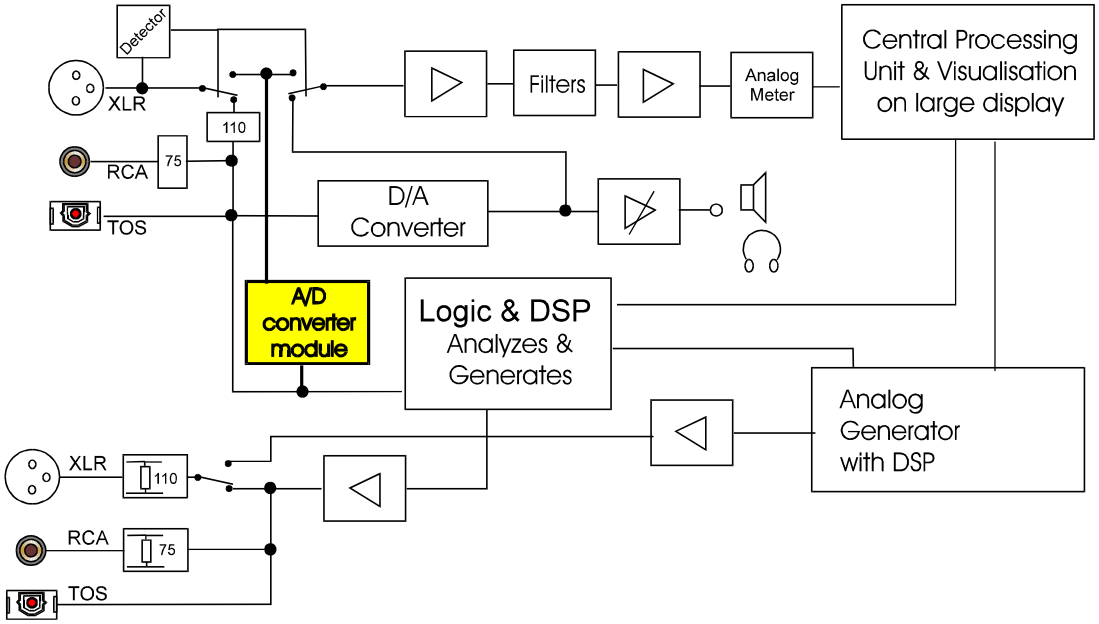


Fig. 26 Block Diagram FFT Option

The A/D converter samples the incoming analog signal in the time domain at a sampling rate of 48kHz. After each block of 2048 samples, the DSP calculates the spectrum from these samples and displays the result in two possible frequency ranges. The low frequency range covers 0Hz-5kHz, while the high frequency range covers the audio band from 0Hz-20kHz.

The calculated FFT spectrum shows the levels of a fixed number of frequencies, being distributed over the X- axis. The scheme, in which these frequencies are arranged, indicates the *Frequency Resolution* of the spectrum, i.e. the frequency spacing *Df*.

$$\Delta f = \frac{f_s}{NoOfSamples} = \frac{48000Hz}{2048} = 23.4Hz$$

Equation 3 Frequency Resolution of FFT

Therefore, the FFT spectrum theoretically displays every 23.4Hz a frequency, i.e. all together a set of 1024 frequency samples up to the Nyquist frequency of $f_s/2$.
Consequently, in the high frequency range (0Hz-20kHz) 856 frequencies should be displayed. However, since the LC display has 214 lines in the graphical area available only, every displayed level actaully represents the maximum value of four neighboured frequencies, while in the low frequency range (0Hz-5kHz), the display resolution is big enough to represent every frequency by an own line.

If a FFT spectrum is printed out, all calculated frequencies are printed, regardless of the selected frequency range, due to the sufficiently high resolution of the printer.

Operation

The FFT display mode is an alternate mode of the SCOPE display and can be opened in the analog mode only.

> ***NOTE*** ***The FFT analysis is available in the analog mode only, for measurement functions LEVEL or 2-CH. Digital signals cannot be generated as long as the FFT display mode is active.***

To enter the FFT mode, first set the measurement function to either LEVEL or 2-CH, and then press the <SCOPE> button. Depending on the actual setup of the A2-D, the input signal will be displayed either the in the time or the frequency domain.

To toggle between these two modes, open the *ScopeSetup* menu by double-clicking to the <SCOPE> button. Therein, either choose *Time Domain* (scope display mode) or *Freq Domain* (FFT display mode) by using the <SCOPE> softwheel. Press the wheel once to confirm.

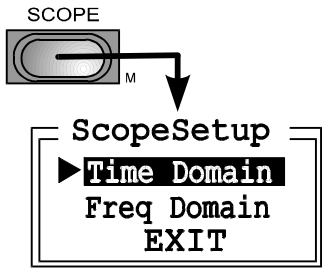


Fig. 27 Scope Setup Menu

In the FFT display mode, any change of the measurement function to another one than 2-CH or LEVEL, switches the display back to the SCOPE mode.

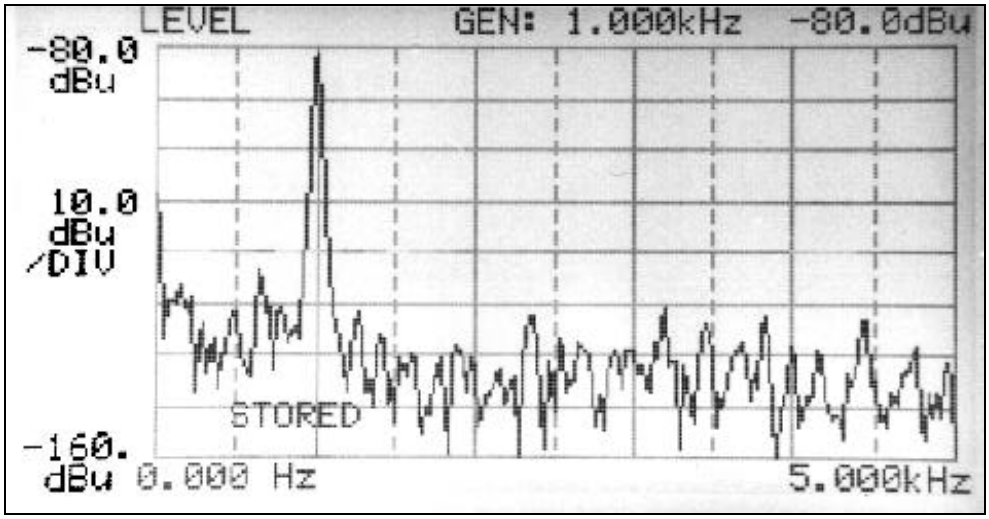


Fig. 28 FFT Display of a Sine Signal

Fig. 28 shows a FFT spectrum in the low frequency band from 0Hz to 5kHz. The X-axis represents the frequency, with each pixel step corresponding to 23.4Hz. The level of the peak signal at 1kHz is -80dBu, representing the RMS value of that single frequency.

Please notice, that the measurement accuracy of the FFT is limited to ≤ 2dB, due to the so-called "picked fence effect".

Frequency Range

The FFT display has two possible frequency ranges, one from 0Hz-5kHz and the other from 0Hz-20kHz.

To select either range, set the generator frequency to a value below or above 5kHz. Next, press the <STOP VALUE> button. The system will then display the low range if the generator frequency is < 5kHz, or the high range if the generator frequency is ≥ 5kHz.

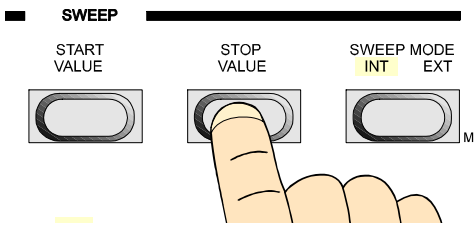


Fig. 29 Frequency Band Selection

> ***NOTE*** ***In the high frequency range (0Hz-20kHz), each screen pixel represents the max. level of four neighboured frequencies, while in the low range mode (0Hz-5kHz) each screen pixel is identical to one frequency.***

Windowing Functions

As required for non-periodic signals, the FFT transformation of the A2-D applies different windowing functions which can be selected through a menu.

In the FFT display mode, double-click to the <SCOPE> button to open the *ScopeSetup* menu. Two of the most commonly used windowing functions, *FlatTop* and *Hanning*, are available. The actually selected windowing function is displayed in inverted letters (white on black). Change the selection by placing the cursor with the <SCALE> wheel and press the wheel to confirm.

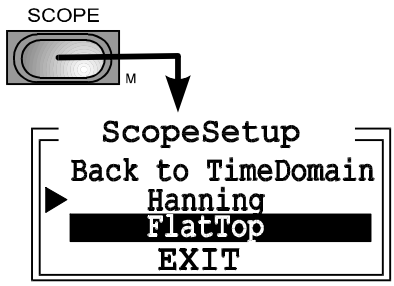


Fig. 30 Windowing Functions

Zoom, Scroll & Autoscale

The FFT display can - like the sweep graph - be zoomed and scrolled. For the operator's convenience, the redrawing process of the spectrum is delayed for a few hundred milliseconds, in order to allow multiple Zoom and Scroll adjustments.

Please notice, that due to the high refresh rate, the <SCALE> wheel is less frequently checked with a stable display than in the redraw mode.

An important difference to the normal graph operation concerns the auto scaling, since the autoranging circuitry uses the RMS level to define the optimum display scale. With signals showing a high crest factor (peak-to-RMS level ratio), such as noise signals, the auto-ranging may fail so that the scale may not optimized or the signal may exceed the display. Re-adjust the scaling in this case by Zooming and Scrolling accordingly.

The FFT display can be frozen at any time by pressing the <SCOPE> wheel once. The next press to the <SCOPE> or any other measurement function or <UNIT> button releases the freeze mode, i.e. the display will be refreshed continuously.

If the instrument is re-ranging its input amplifiers, the message *Ranging* will be displayed in the top right corner of the display. During that period the system is not able to display any spectrum.

Printouts

Every FFT spectrum can be printed out by pressing the <PRINT> button. The printout is handled in parallel to the FFT calculation, so that the display will be refreshed continuosly even during the printout process.

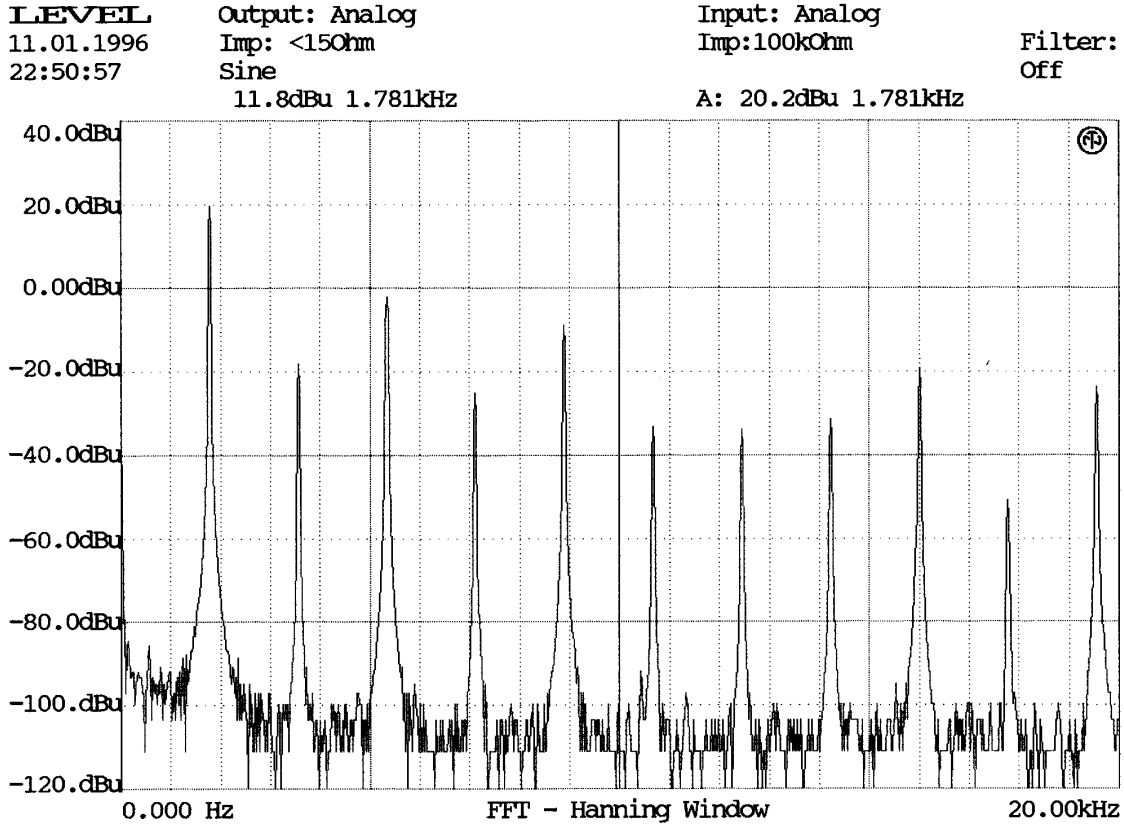


Fig. 31 Printout of FFT Display

> ***NOTE*** ***The resolution of a printout is increased compared to the screen resolution in the high range mode.***

Leaving the FFT Display Mode

There are different possibilities to leave the FFT display mode.

- Enter the METER or GRAPH display mode by pressing the corresponding key.
- Open the *ScopeSetup* menu (see ***Fig. 30***) and select the *Back to TimeDomain* entry.
- Activate a measurement function other than LEVEL or 2-CH.

7 AES/EBU DEFINITIONS

The demand for digital signal transmission in the professional audio industry forced the AES to release a standard for the standardized transmission of such signals. This is the AES-3 (or AES/EBU) standard. In parallel the earlier de-facto standard introduced from SONY and PHILIPS for Digital Interface (SP-DIF) has also been standardized by IEC958 for unbalanced consumer links.
The main differences of the physical transmission between the AES/EBU and IEC958 standard are level, impedance of the cable and the type of connection.
The transmission standards are - beside the mentioned differences - more or less identical, expect that the AES/EBU standard defines only the professional format, while the IEC standard covers both the professional and the consumer format. The differences between these format will be explained later.
Since Audio signals are in most applications two-channel for stereo reproduction, the standards foresee the serial transmission of both channels one after the other, completed with additional information.

Definition

The audio data of each channel are packed into 32bit-long *subframes*, that start with a 4bit synchronization *preamble*, followed by 4bits *auxiliary* data and 20 *Audio* data bits (LSB first). If less than the max. of 24bits (4 auxiliary + 20 audio) are required, the unused bits are set to zero. Each subframe is terminated by the *Validity* flag, a *User* bit, the *Channel Status* and the *Parity* bit. Two subframes, one for each channel, build together a *frame*.

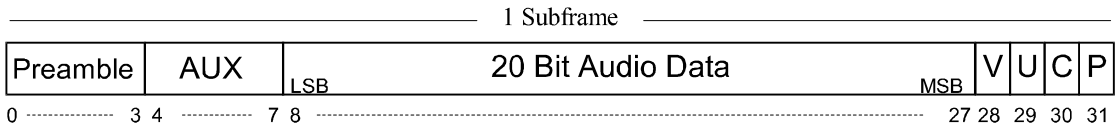


Fig. 32 AES-3 Subframe

- Please notice, that alternatively to the extension of the Audio data to 24bits, the auxiliary data can also be used to transmit any other information.
- The validity flag expresses, whether the Audio sample in that word is valid or not.
- The user bit can be set freely, while there exists an AES recommendation (AES18 / ANSI S4.52-1992) for the format of the user bit.

192 frames are always grouped together to a block. The meaning of the status bit has to be seen in the larger block contents, where these status bits are grouped together to 192 bits, whereof some are interpreted as Bytes. A detailed explanation follows. The parity bit builds a checksum over the subframes to find single bit errors.

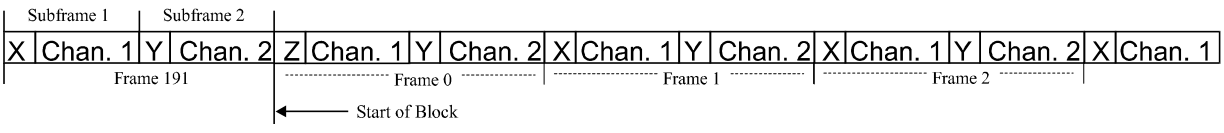


Fig. 33 Start of Block with Preamble

One block of consists of 192 frames as illustrated above. Therein, each subframe is headed by an individual preamble, whereof the **X**-preamble marks the A-channel and the **Y**-preamble the B channel. A special preamble, the **Z**-preamble, replaces at each beginning of a block

the **X**-preamble once to indicate the start of a new block. This allows the receiving devices to trigger to the data stream and to find the beginning of a block, which is evident for the proper interpretation of the signal.

Signal Transmission

The signal transmission is performed in the so-called bi-phase mark mode. This mode has been chosen to minimize the DC-component in the physically transmitted signal. The illustration below shows, that a logical 1 is represented by a polarity change in the middle of the source signal, while a logical 0 does not show this transition.
Please notice, that the clock of the digital bitstream therefore has to be twice the sampling rate times 64 (2 channels with 32 bits) which results in a clock of ~6MHz at a given sampling rate of 48kHz.

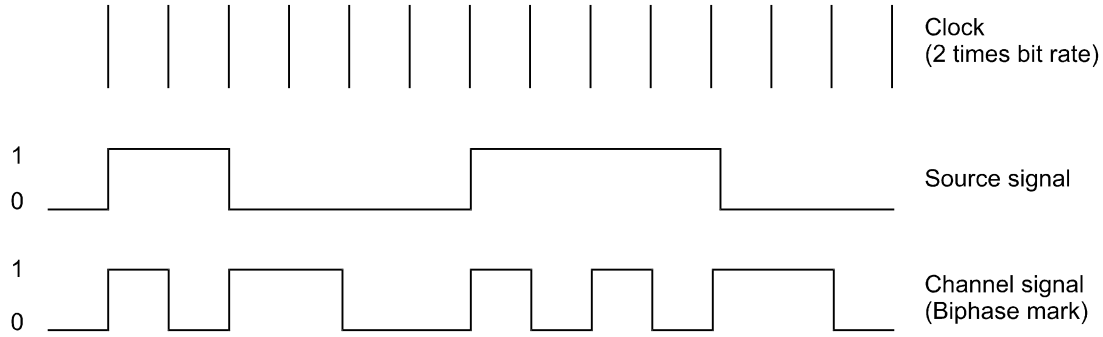


Fig. 34 Bi-phase Signal Coding

Channel Status Information

Every block of the transmission contains 192 frames with one Channel 1 subframe and one Channel 2 subframe each. Consequently, since each subframe contains one channel status bit, 192 channel status bits per channel are transmitted with every block. Obviously, with the start of the next block, this status information is repeated.
In practice, the 192 status bits are grouped together into Bytes (8bits), to represent ASCII characters. The meaning of the bits and bytes in the channel status are different for professional and consumer format. Actually, only the bit 1and bit 2 of the status are identical to ensure proper decoding of the professional and consumer format.

Professional Format

The following illustration shows the allocation of the channel status block for professional purposes.

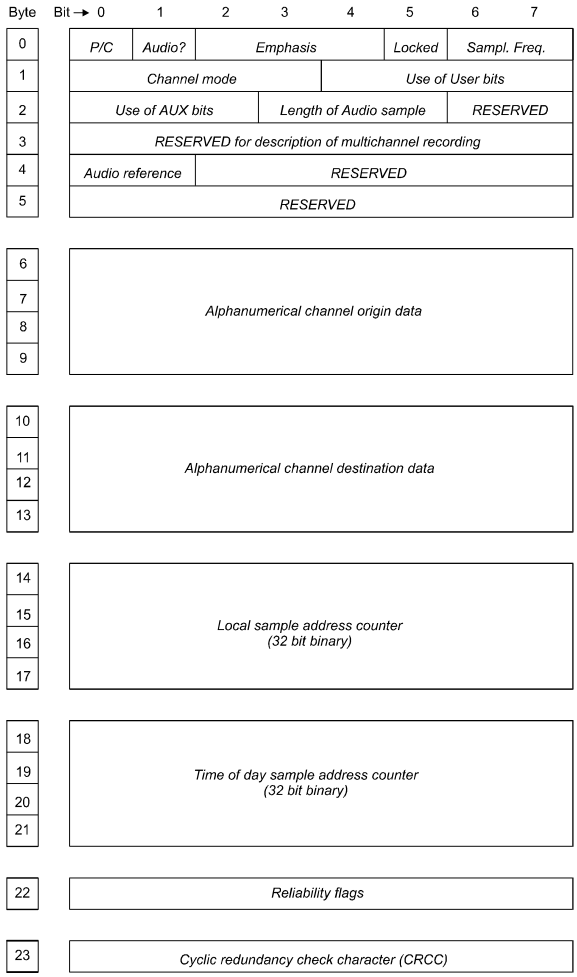


Fig. 35 PRO Channel Status

Byte 0

The first bit designates the *format*. A logical 1 states the professional format, while 0 represents consumer format. Bit 2 indicates whether the transmitted data are *audio* data. The three *emphasis* bits define the type of emphasis used. Of the eight possibilities, only 4 are used and the others are reserved.
Bit 5 indicates whether the source sampling frequency has been locked or not. If this bit is *unlock* then the Audio data is not reliable and my contain garbage. The following bits six and seven contain the information about the *sampling rate*. 32kHz, 44.1kHz, 48kHz and no-indication are defined. The sampling rates are not measured but set by the transmitter.

Byte 1

The bits 0 to 3 encode the *relation* of the two channels. It ranges from not indicated to two channel, single channel and stereophonic. All further combinations are reserved for future applications.
The second four bits are reserved for the *user bits management*, defining the default mode, the Z-preamble mode and a user-defined application. All others are reserved.

Byte 2

Byte 2 defines the use of the auxiliary bits and the *length* of the Audio data. The first three bits code the maximum length of the Audio data to either Not defined, 24 bits, 20bits, or user defined. Bits 3 to 5 designate how long the Audio data in the previously defined frame really are. Reductions by 0, 1, 2, 3 and 4 bits are defined. The bits 6 and 7 are reserved.

Byte 3

Byte 3 is completely reserved for future application.

Byte 4

The first two bits define whether the audio signal is a reference signal and of what *type*, while the rest of the bits in that byte are reserved.

Byte 5

Completely reserved for future applications.

Byte 6 to 9

Theses bytes are interpreted as a set and represent the alphanumeric *channel origin* data. They are normally represented as ASCII characters. There exists no restriction, what character code is generated by any device. Examples may be CD, RDAT, GEN or A2.

Bytes 10-13

Are again ASCII data representing the *channel destination* data. The same restriction as for the channel origin data may be applied.

Bytes 14-17

Is an ongoing sample *counter* with 32bits width. Even though the counter increases by 48000 units per second, the maximum number that can be represented is 2^{32} = 4'294'967'296. The counter restarts at zero approximately every 25 hours.

Bytes 18 to 21

The bytes represents in 32bit format the *time* of day sample address. This allows, with the knowledge of the sampling address, to easily calculate the time difference between any frames.

Byte 22

This byte carries information about the *confidence* of the data in the channel status bits. The first four bits are again reserved. Byte four represents an uncertainties exists in the bytes 0 to 5, while the 1 in bit five indicates a possible error in the bytes 6 to 13. Bit six states an unreliability in the bytes 14 to 17. A problem in the bits 18 to 21 is then signalized in bit 7.

Byte 23

Contains a 32 bit *checksum* (CRC) over all channel status information to indicate possible bit changes during transmission. The polynomial is: $G(x) = X^8+X^4+X^3+X^2+x+1$.

Consumer Format

The consumer was formed out of the earlier SONY and PHILIPS standard. These companies worried most about the copy protection, while the discussion is still not finished, how the copy flag shall behave after the first or second copy through the digital output.

The consumer status block defines much less information than the professional format. Only the first four bytes are defined, the rest is unused.

Byte 0

The use of the first bit is identical with the professional format to ensure homogeneous interpretation of the formats.
Bit one defines Audio or non-audio use. Bit 2 is - already different from the professional format - the *copy protection* bit. With this bit being zero, no copy of the audio material is allowed.
Bits 3 to 5 define the type of Audio data while bit 6 and 7 always are set to zero. The scheme for the copy protection is not yet fixed but still under development.

Byte 1

The first 3 bits define the *category* code. Depending on the category, the interpretation of the next bits in that byte is completely different.

Byte2

This byte defines in its first four bits the *source number* with the LSB first.
The next four bits define the channel number from A to O.

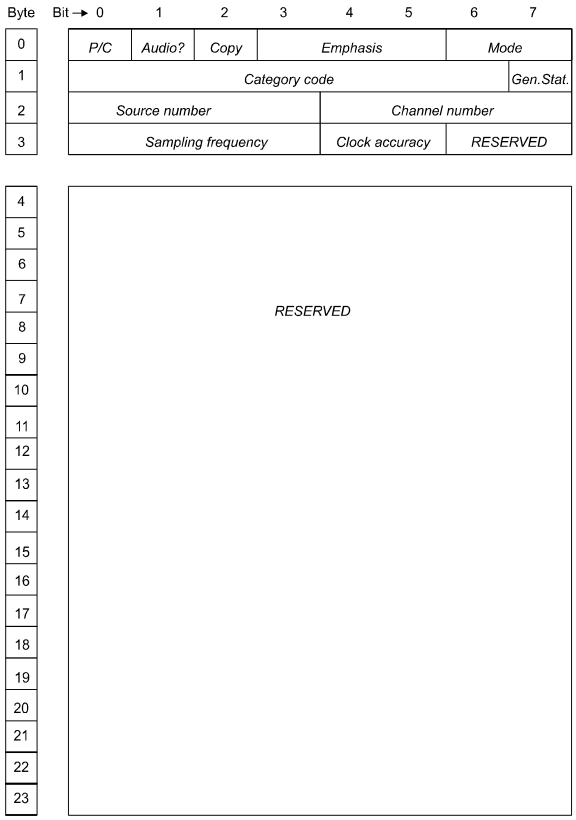


Fig. 36 Consumer Channel Status

Byte 3

Decodes the used sample rate in the first four bits. Bit 4 and 5 define the accuracy of the clock, while bit 6 and 7 are reserved.

Byte 4-23

The content of these bytes is reserved for future use.

Signal Transmission

Besides the status and audio information, there also exist major differences between the physical signal transmission between AES/EBU and the IEC958 signal. AES defines a balanced transmission, while the IEC standard defines an unbalanced electrical or optical link.

Balanced Transmission

The balanced transmission operates with 110Ω impedance, while the electrical parameters are defined to allow a signal transmission up to a few hundred meters. The acceptable carrier level has to be between 0.2Vpp and 5Vpp. The lower limit was earlier defined at 2Vpp, but has been adjusted to accept also signals coming from an unbalanced transmission without any re-amplification.

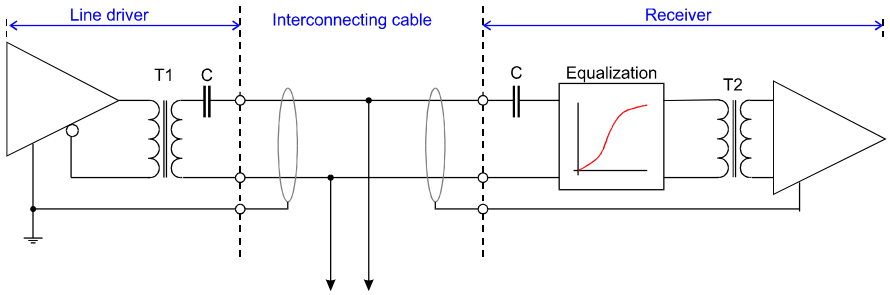


Fig. 37 Balanced Connection of AES-3 Signals

Please notice, that for good and reliable RF-shielding both ends of the screen have to be connected - in opposition to the practice of analog audio connections, where ground loops have to be avoided.

Unbalanced Transmission

The IEC standard recommends an unbalanced or optical link, allowing transmission lengths of up to ten meters.

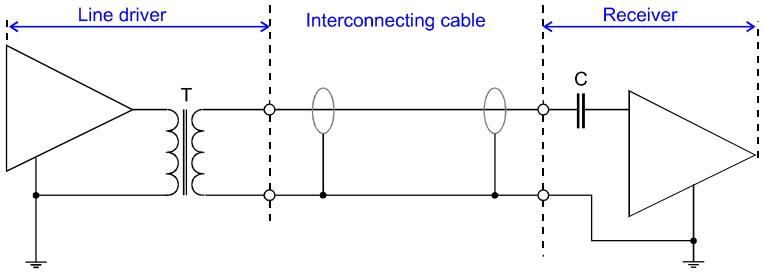


Fig. 38 IEC958 Unbalanced Connection

Eye Pattern Test

The eye pattern test is the only standardized Audio quality test in the AES standard. It requires an analog oscilloscope with an external trigger input. The interpretation of the results needs some additional knowledge and calculation.

Every transmission line has a low-pass characteristics, deforming the shape of an ideal square signal. Time instability additionally varies the zero crossing of the signal.

An ideal square signal is deformed to the lowpass filtered signal. Trigger is synchronous to the positive edge of the square signal. The trace re-starts with the positive slope. The scope is then again re-triggered with the negative edge of the square signal, to draw the trace of the negative shape. If the beam density is high enough the screen will show several of the traces, forming one half of the oval eye pattern.
The opening of the eye indicates the level attenuation and degradation, while the with of the eye is influenced by the 1 and 0 as well as by jitter. The eye height shall be high enough to let the input circuitry safely decide between a logical zero and a logical one. Below a certain hysteresis the input device won't be able to make the decision and starts reading errors. This is the minimum opening height.

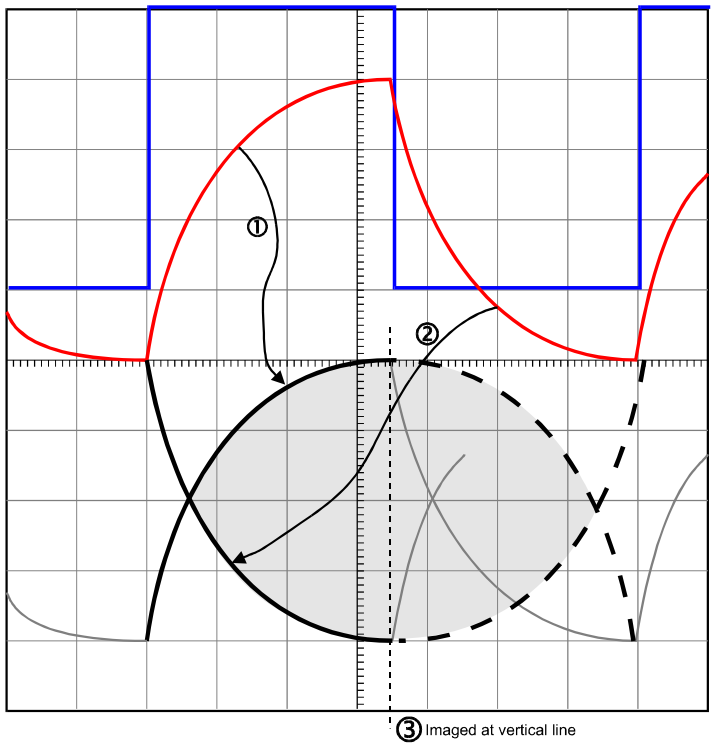


Fig. 39 Construction of Eye Diagram

The AES-3 standard specifies the minimum voltage to be bigger than 200mV for at least 50% of the nominal eye width.

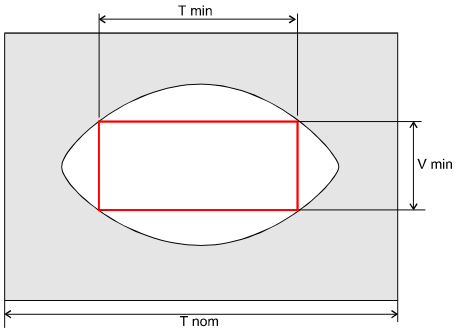


Fig. 40 Specs of the Eye Pattern

8 TECHNICAL SPECIFICATIONS

Inputs

AES/EBU through XLR connectors at front panel with 110Ω input impedance

IEC958 through RCA and optical TOSLINK at rear panel with 75Ω input impedance

Sample clock through BNC at rear panel.

Sampling frequency	32kHz, 44.1kHz, 48kHz or external
External clock	25kHz to 52kHz (locking range)

Measurement Functions

Digital Measurements

Peak Level of Data Carrier	100mVpp - 10Vpp
Clock Frequency	25kHz - 50kHz
Frequency Resolution	0.05Hz
Frequency Accuracy	2ppm calibrated

Bit statistics of complete 32 bit word

Status display with decoded information for AES and IEC format

Word Clock Jitter

Amplitude	0 - 40ns
Rectifier	Peak
Bandwidth	700Hz - 20kHz

Level Measurement

Range	-120dBF - 0dBF (0 - 100%)

D/A Converted Signal Measurements

RMS Level / THD+N	RMS calibrated to 0dBr with 0dBF (Sine) see analog specifications of A2
Resolution of D/A Converter	-94dB

Mixed Signals

Analog generation and digital analysis
Digital generation and analog analysis

Outputs

AES/EBU at XLR connector channel A on front panel, 110Ω output impedance

IEC958 through RCA and optical TOSLINK connectors at the rear panel with 75Ω output impedance.

X/Y/Z trigger output through BNC connector at the rear panel

Sampling Frequency	32kHz, 44.1kHz, 48kHz or input clock or analyzer clock
Accuracy	10ppm (2ppm on request)
De-tuning of fs	±1500ppm in steps of 250ppm
Carrier Output Level	100mVpp - 5Vpp (loaded into 110Ω)
Output Signals	Sine or Sine + Dither or Pass or Pass + Dither or CRC Test Signal
Frequency Range	10Hz to $f_s/2$
Level Range	-120dBF - 0 dBF (0% - 100%)
Resolution of Audio Data	4, 8, 12, 16, 18, 20, 24bit
Dithering	1 LSB triangular probability density function
Internal Jitter Generator	2ns - 40ns (-84dBUI to -54dBUI) @ 48kHz
Modulation Signal	Sinusoidal 2.5kHz or White noise with 20Hz - 20kHz bandwidth
Output Impedance	110Ω for AES/EBU signals 75Ω for IEC958 signals

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